US009060037B2

(12) **United States Patent**
Mang et al.

(10) **Patent No.: US 9,060,037 B2**
(45) **Date of Patent: *Jun. 16, 2015**

(54) **METHOD AND APPARATUS FOR PRIORITIZING VOICE OVER INTERNET PROTOCOL SIGNALING MESSAGES**

(75) Inventors: **Xiaowen Mang**, Morganville, NJ (US); **Carolyn Roche Johnson**, Holmdel, NJ (US)

(73) Assignee: **AT&T Intellectual Property I, L.P.**, Atlanta, GA (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 134 days.

This patent is subject to a terminal disclaimer.

(21) Appl. No.: **13/620,221**

(22) Filed: **Sep. 14, 2012**

(65) **Prior Publication Data**

US 2013/0016611 A1 Jan. 17, 2013

Related U.S. Application Data

(63) Continuation of application No. 12/210,701, filed on Sep. 15, 2008, now Pat. No. 8,305,890.

(51) **Int. Cl.**

H04L 29/08	(2006.01)
H04M 3/36	(2006.01)
H04L 29/06	(2006.01)

(52) **U.S. Cl.**

CPC ***H04L 67/322*** (2013.01); ***H04M 3/367*** (2013.01); ***H04L 65/1006*** (2013.01); ***H04L 65/1033*** (2013.01)

(58) **Field of Classification Search**

None

See application file for complete search history.

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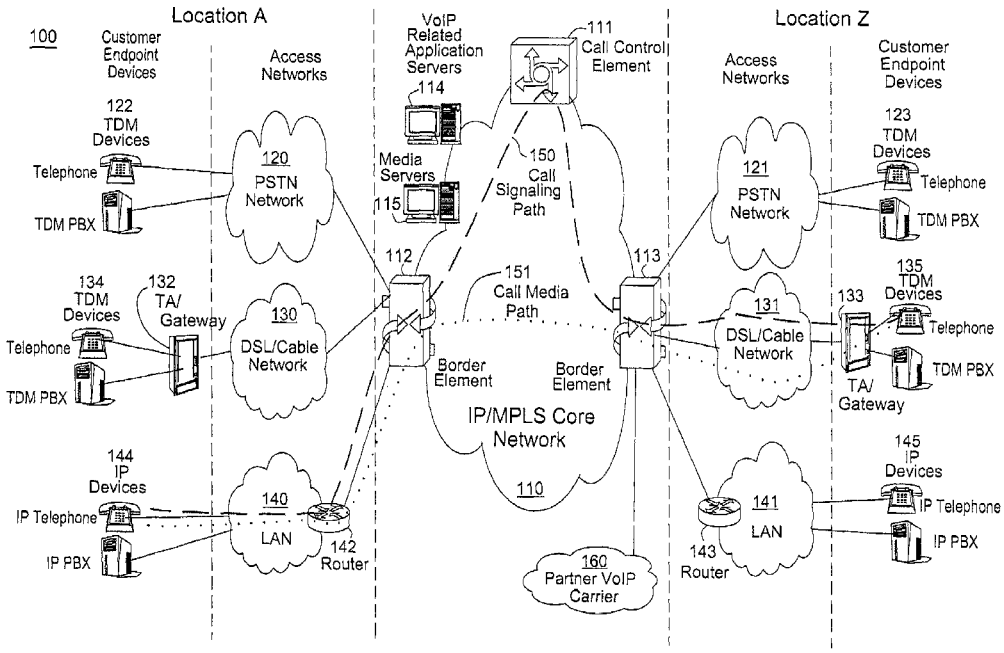
Primary Examiner — Gregory Sefcheck
Assistant Examiner — Daniel Mitchell

(57) **ABSTRACT**

A method and apparatus for enabling prioritization of signaling messages in a communication network are disclosed. For example, the method receives at least one signaling message, and classifies each of the at least one signaling message. The method schedules each of the at least one signaling message for processing, and discards selectively one or more signaling messages that have been scheduled under an overload condition.

14 Claims, 12 Drawing Sheets



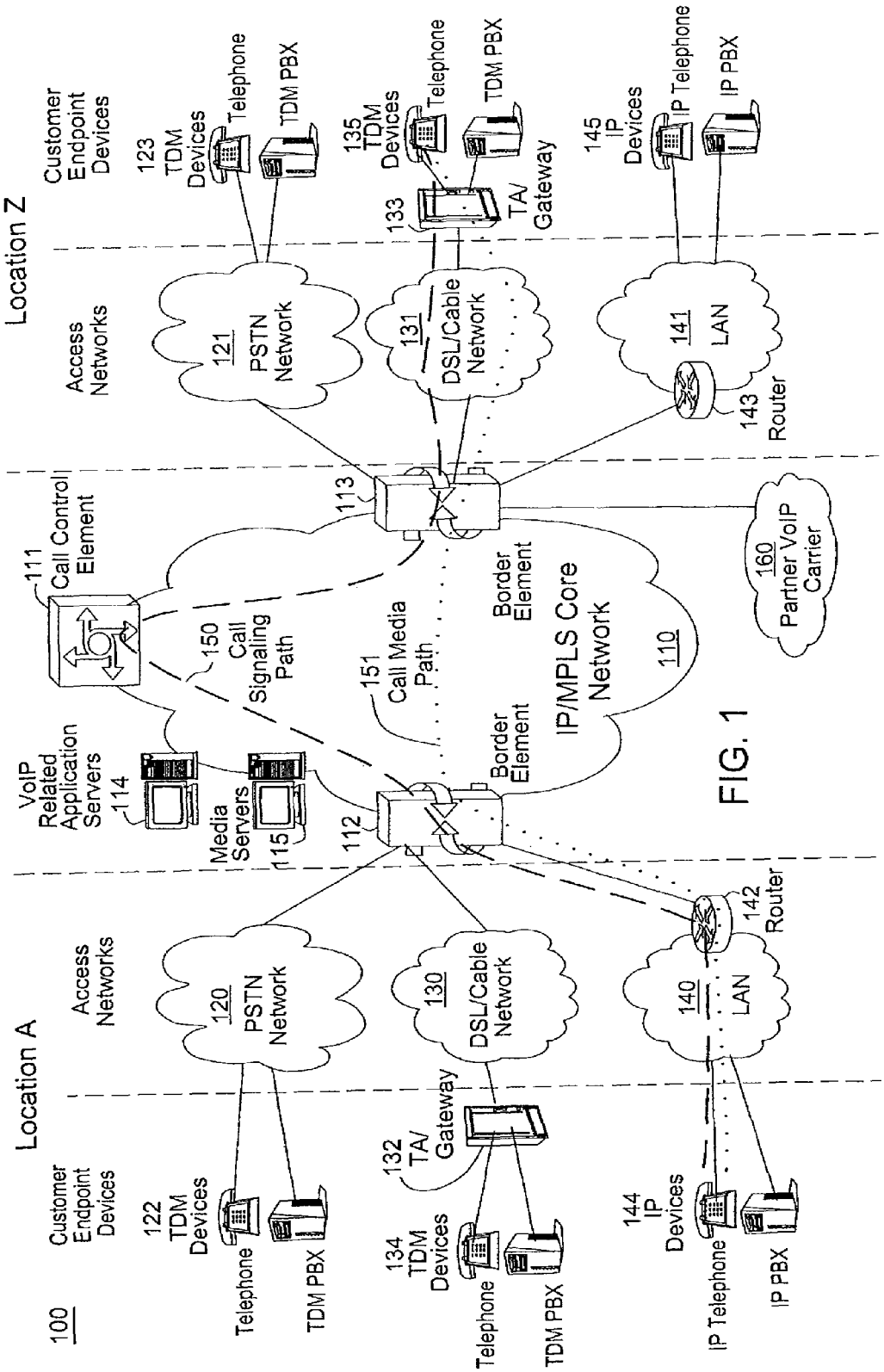

100
Location A
Customer Endpoint Devices
Access Networks
122 TDM Devices
Telephone
TDM PBX
120 PSTN Network
130 DSL/Cable Network
132 TA/ Gateway
134 TDM Devices
Telephone
TDM PBX
140 LAN
142 Router
144 IP Devices
IP Telephone
IP PBX
VoIP Related Application Servers 114
Media Servers 115
112
111 Call Control Element
150 Call Signaling Path
151 Call Media Path
Border Element
Border Element
IP/MPLS Core Network 110
113
160 Partner VoIP Carrier
Location Z
Customer Endpoint Devices
Access Networks
123 TDM Devices
Telephone
TDM PBX
121 PSTN Network
131 DSL/Cable Network
133 TA/ Gateway
135 TDM Devices
Telephone
TDM PBX
141 LAN
143 Router
145 IP Devices
IP Telephone
IP PBX

FIG. 1

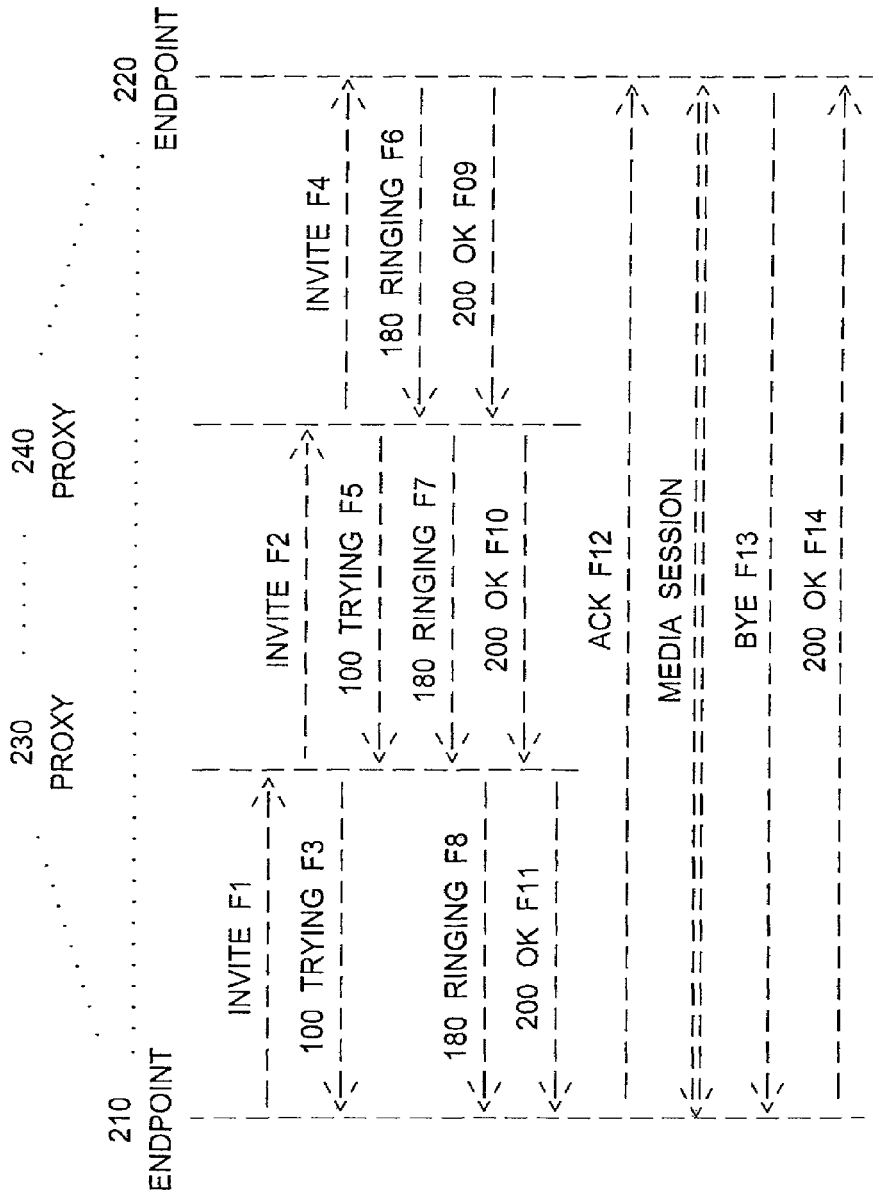

210
ENDPOINT
230
PROXY
240
PROXY
220
ENDPOINT
INVITE F1
100 TRYING F3
INVITE F2
100 TRYING F5
INVITE F4
180 RINGING F6
180 RINGING F8
180 RINGING F7
200 OK F11
200 OK F10
200 OK F09
ACK F12
MEDIA SESSION
BYE F13
200 OK F14
200

FIG. 2

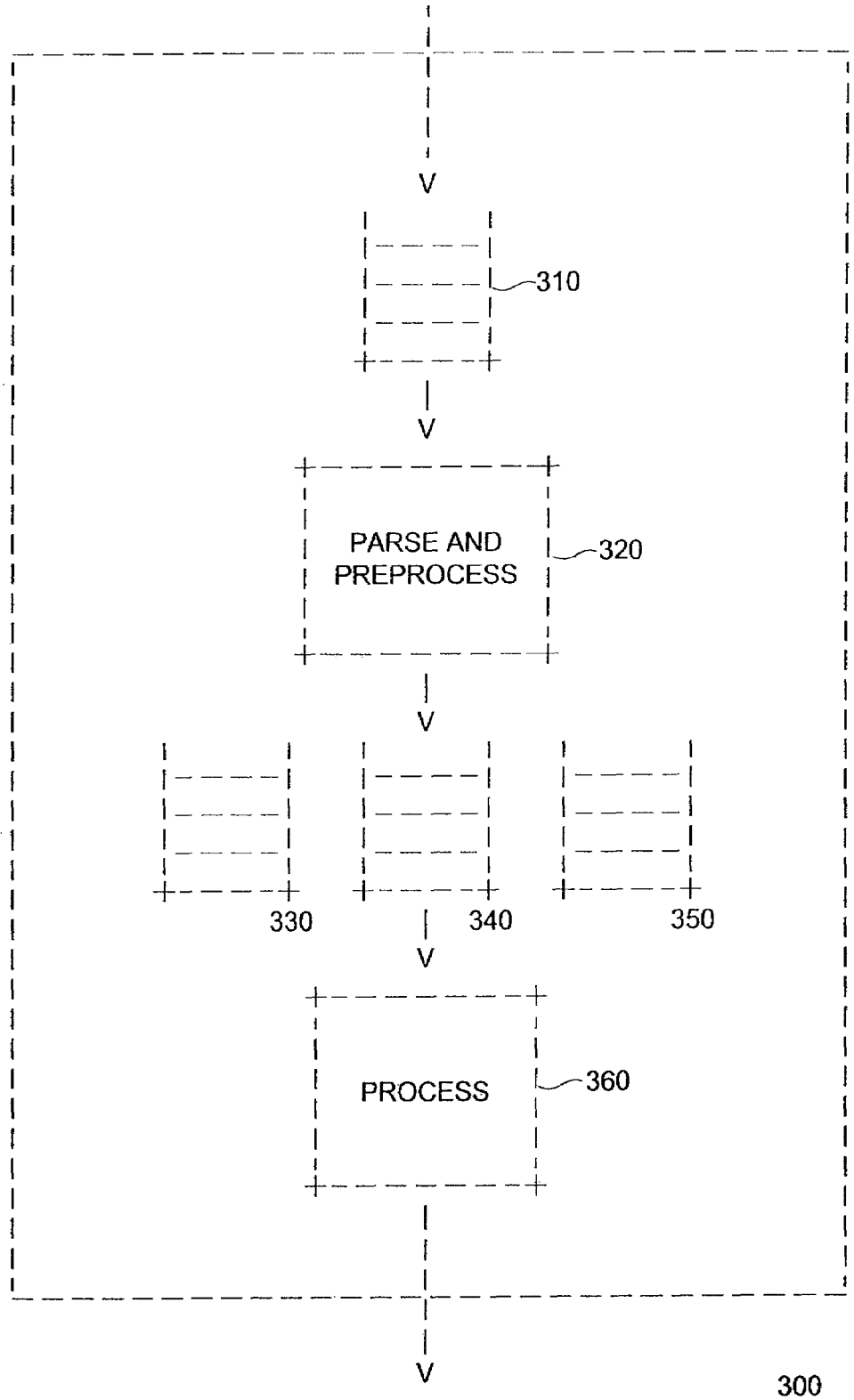

V
310
V
PARSE AND PREPROCESS
320
V
330
340
350
V
PROCESS
360
V
300

FIG. 3

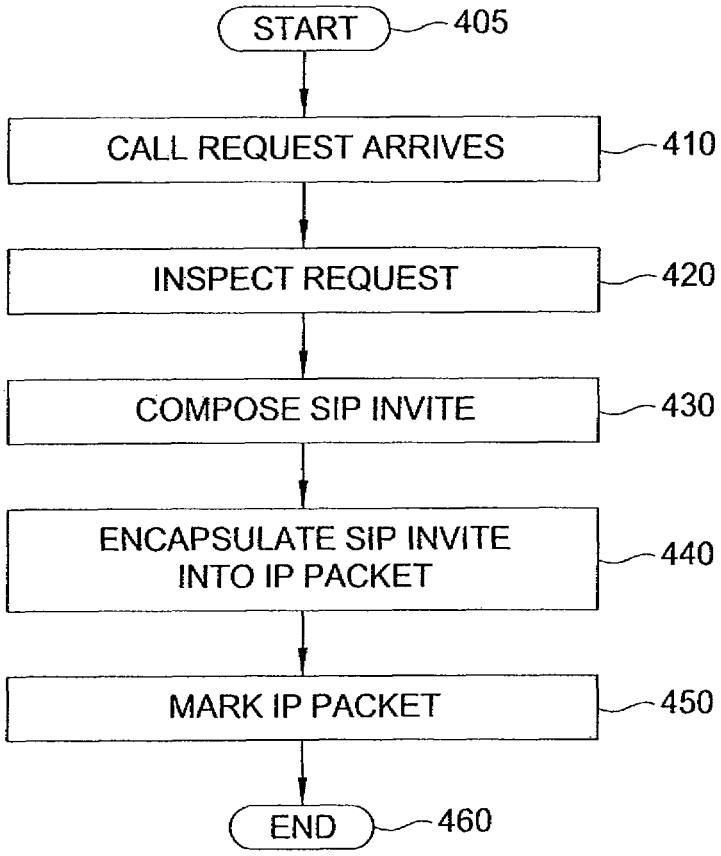
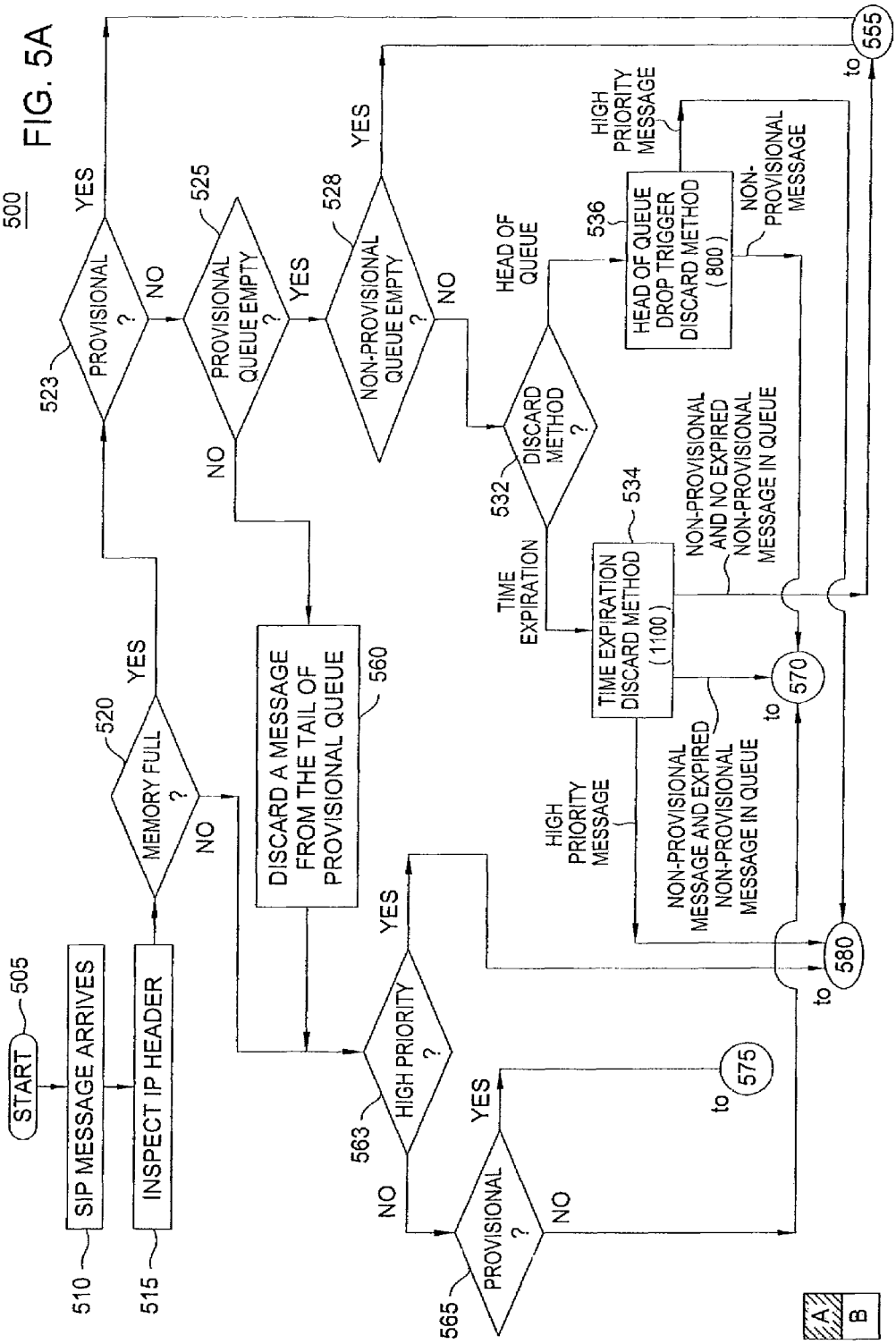

START
405
CALL REQUEST ARRIVES
410
INSPECT REQUEST
420
COMPOSE SIP INVITE
430
ENCAPSULATE SIP INVITE INTO IP PACKET
440
MARK IP PACKET
450
END
460
400

FIG. 4

FIG. 5A


500
START
505
SIP MESSAGE ARRIVES
510
INSPECT IP HEADER
515
MEMORY FULL ?
520
YES
NO
DISCARD A MESSAGE FROM THE TAIL OF PROVISIONAL QUEUE
560
PROVISIONAL ?
523
YES
NO
PROVISIONAL QUEUE EMPTY ?
525
YES
NO
NON-PROVISIONAL QUEUE EMPTY ?
528
YES
NO
DISCARD METHOD ?
532
HEAD OF QUEUE
TIME EXPIRATION
HEAD OF QUEUE DROP TRIGGER DISCARD METHOD (800)
536
TIME EXPIRATION DISCARD METHOD (1100)
534
HIGH PRIORITY MESSAGE
NON-PROVISIONAL MESSAGE
NON-PROVISIONAL AND NO EXPIRED NON-PROVISIONAL MESSAGE IN QUEUE
NON-PROVISIONAL MESSAGE AND EXPIRED NON-PROVISIONAL MESSAGE IN QUEUE
HIGH PRIORITY ?
563
YES
NO
PROVISIONAL ?
565
YES
NO
to 555
to 570
to 580
to 575
A
B

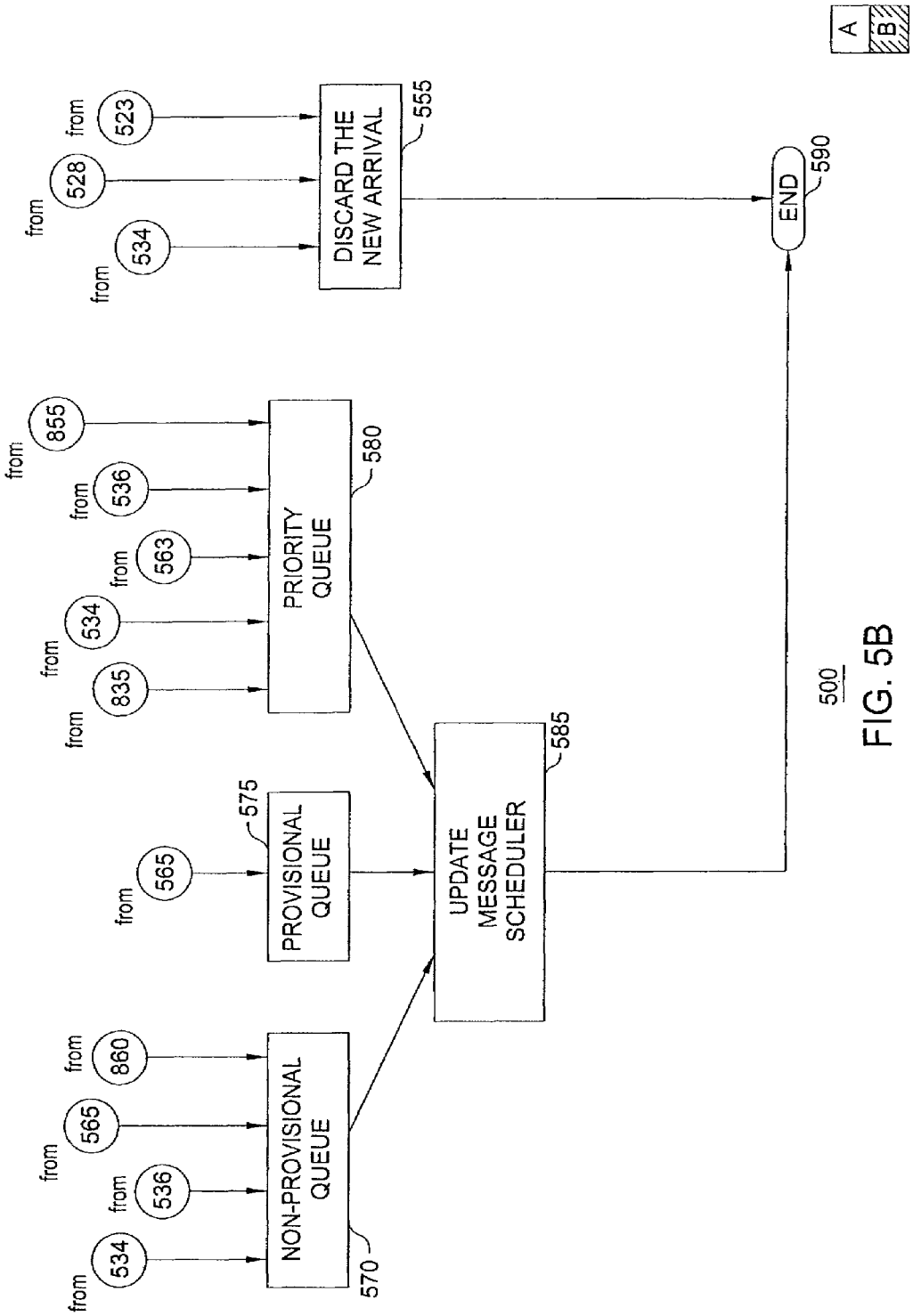

A
B
DISCARD THE NEW ARRIVAL
555
from 523
from 528
from 534
END
590
PRIORITY QUEUE
580
from 855
from 536
from 563
from 534
from 835
PROVISIONAL QUEUE
575
from 565
UPDATE MESSAGE SCHEDULER
585
NON-PROVISIONAL QUEUE
570
from 860
from 565
from 536
from 534
500

FIG. 5B

FIG. 6

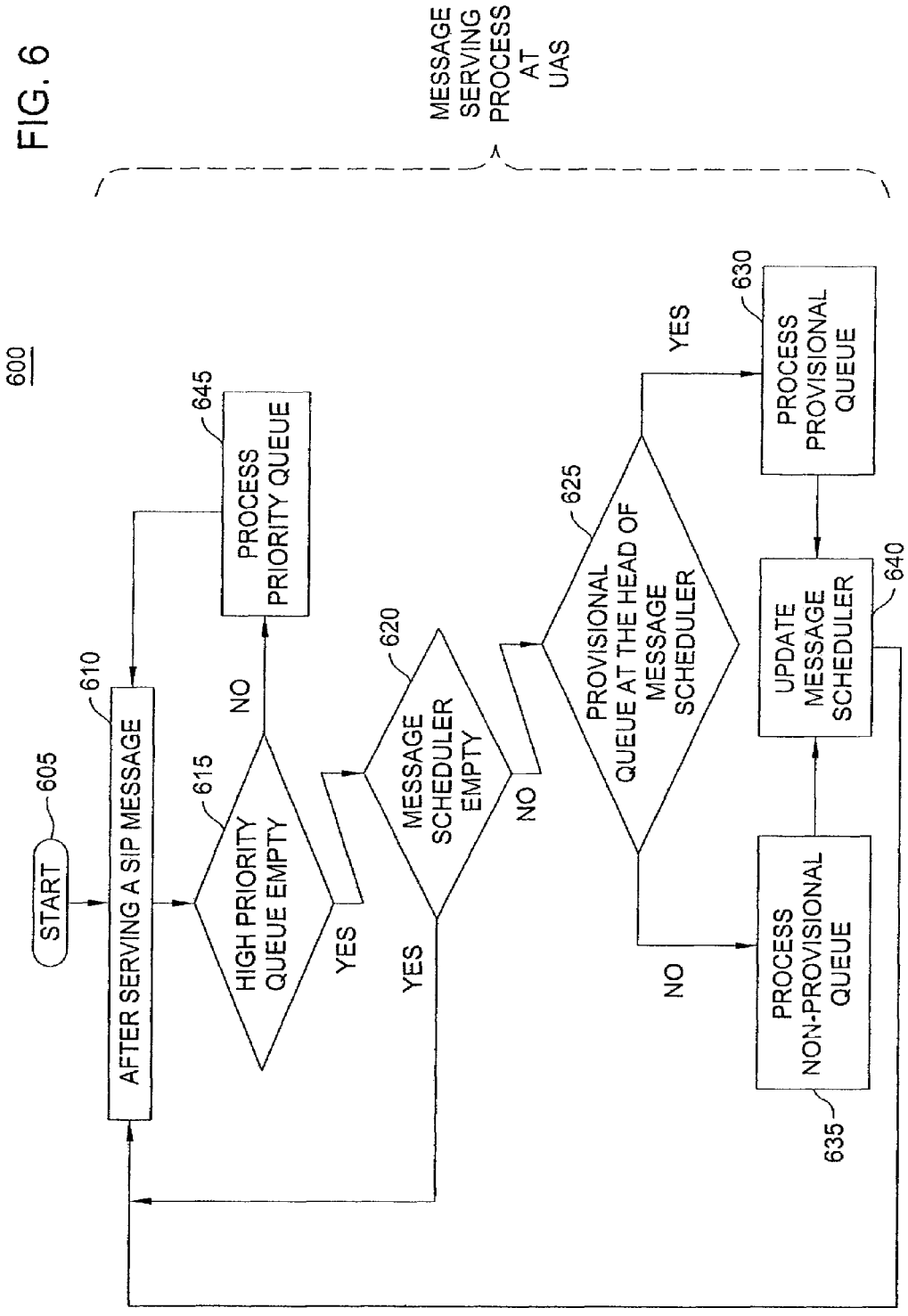

600
MESSAGE SERVING PROCESS AT UAS
START
605
AFTER SERVING A SIP MESSAGE
610
HIGH PRIORITY QUEUE EMPTY
615
NO
YES
PROCESS PRIORITY QUEUE
645
MESSAGE SCHEDULER EMPTY
620
YES
NO
PROVISIONAL QUEUE AT THE HEAD OF MESSAGE SCHEDULER
625
YES
NO
PROCESS PROVISIONAL QUEUE
630
PROCESS NON-PROVISIONAL QUEUE
635
UPDATE MESSAGE SCHEDULER
640

FIG. 7

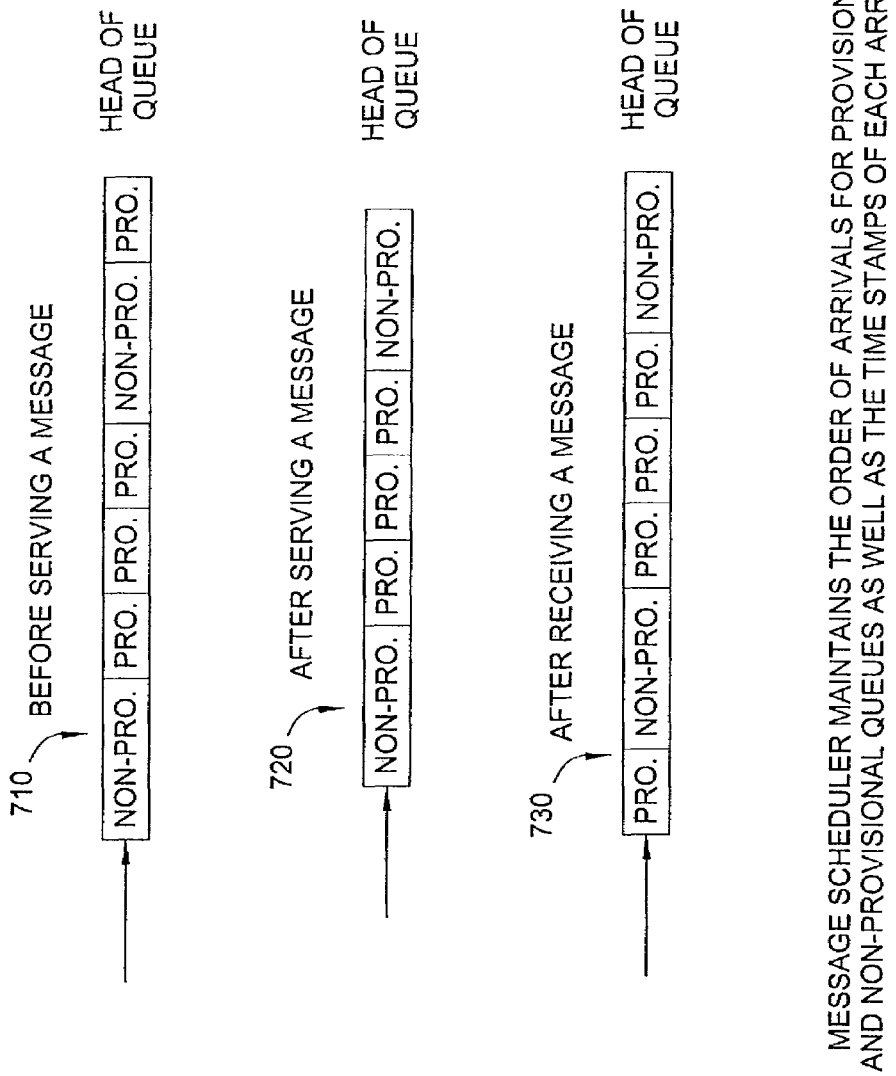

0
710
BEFORE SERVING A MESSAGE
NON-PRO.
PRO.
PRO.
NON-PRO.
PRO.
HEAD OF QUEUE
720
AFTER SERVING A MESSAGE
NON-PRO.
PRO.
PRO.
NON-PRO.
HEAD OF QUEUE
730
AFTER RECEIVING A MESSAGE
PRO.
NON-PRO.
PRO.
PRO.
NON-PRO.
HEAD OF QUEUE
MESSAGE SCHEDULER MAINTAINS THE ORDER OF ARRIVALS FOR PROVISIONAL AND NON-PROVISIONAL QUEUES AS WELL AS THE TIME STAMPS OF EACH ARRIVAL

FIG. 8

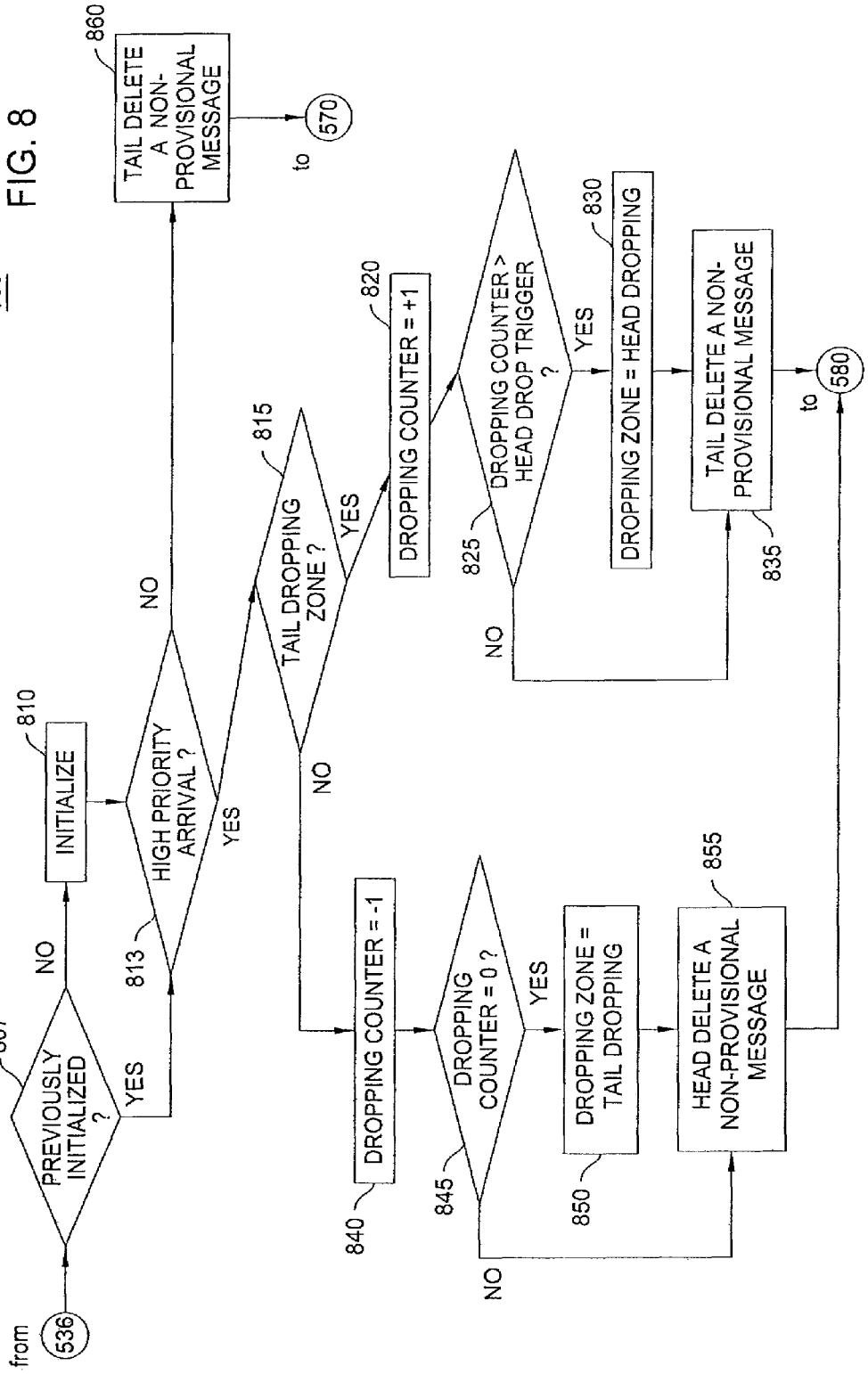

800
from
536
PREVIOUSLY INITIALIZED ?
807
NO
YES
INITIALIZE
810
HIGH PRIORITY ARRIVAL ?
813
NO
YES
TAIL DROPPING ZONE ?
815
NO
YES
DROPPING COUNTER = -1
840
DROPPING COUNTER = 0 ?
845
NO
YES
DROPPING ZONE = TAIL DROPPING
850
HEAD DELETE A NON-PROVISIONAL MESSAGE
855
DROPPING COUNTER = +1
820
DROPPING COUNTER > HEAD DROP TRIGGER ?
825
NO
YES
DROPPING ZONE = HEAD DROPPING
830
TAIL DELETE A NON-PROVISIONAL MESSAGE
835
to
580
TAIL DELETE A NON-PROVISIONAL MESSAGE
860
to
570

FIG. 9

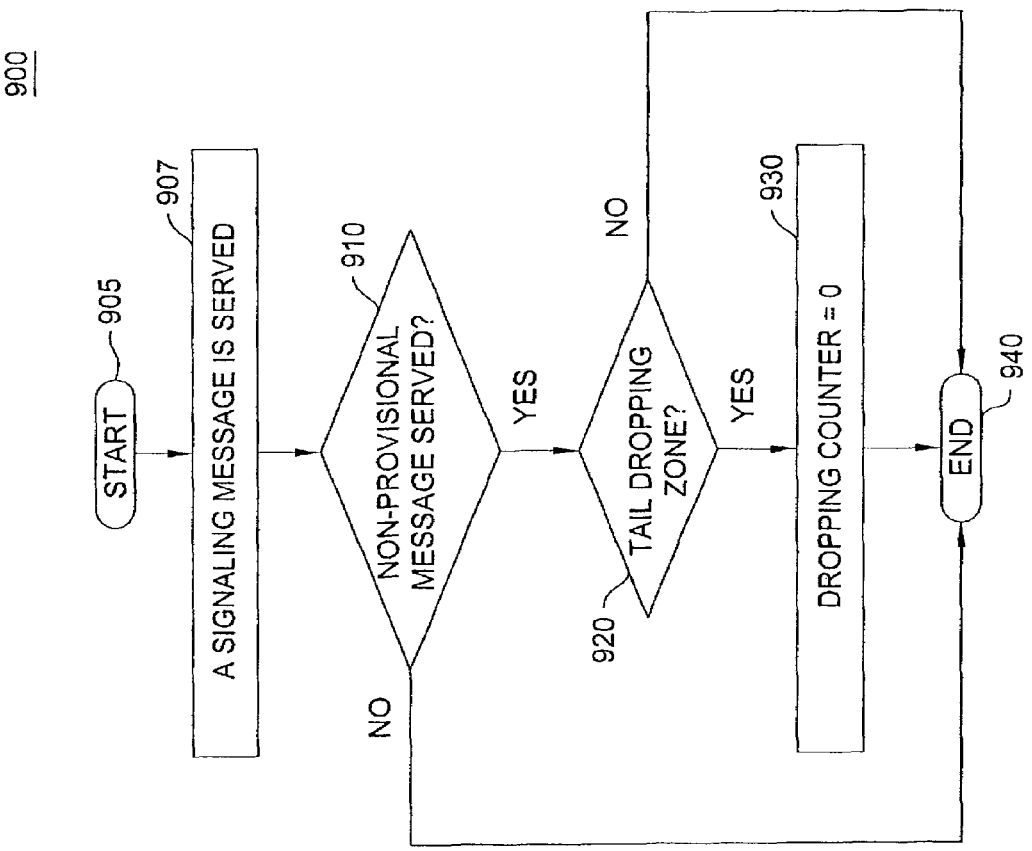
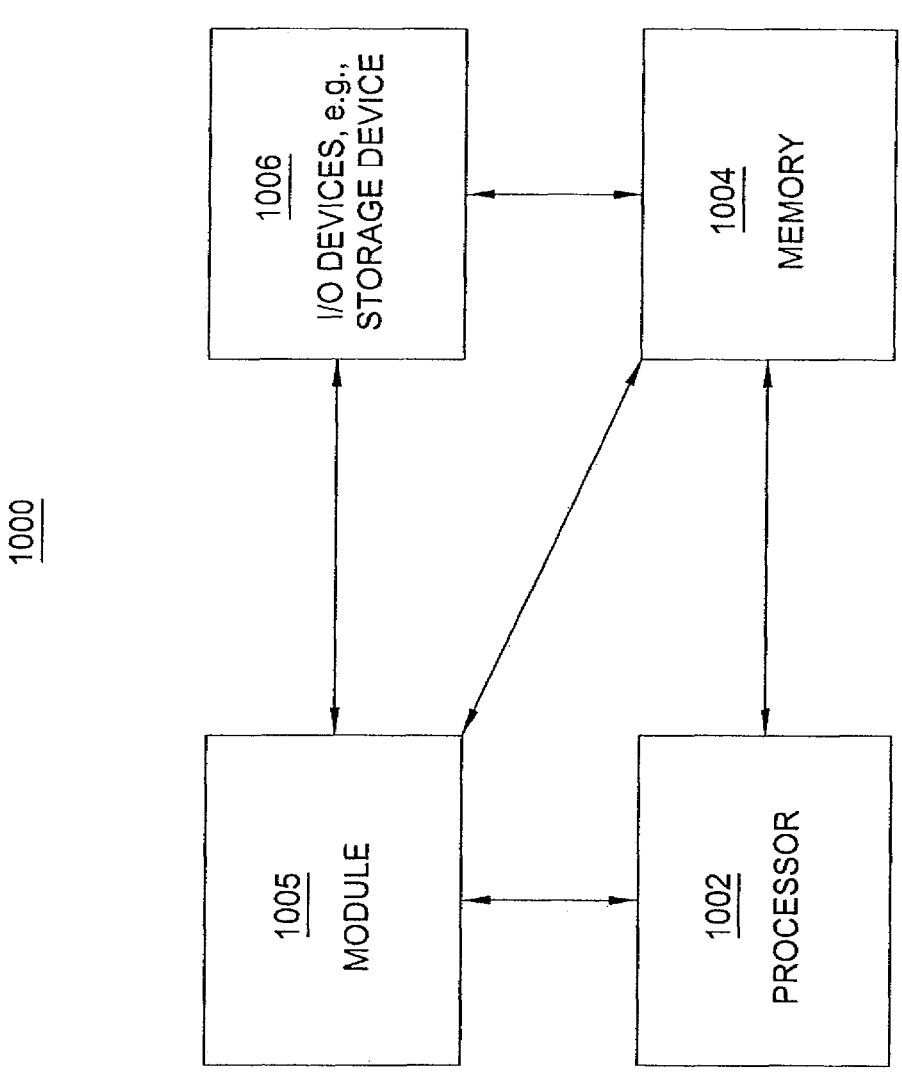

900
START
905
A SIGNALING MESSAGE IS SERVED
907
NON-PROVISIONAL MESSAGE SERVED?
910
YES
NO
TAIL DROPPING ZONE?
920
YES
NO
DROPPING COUNTER = 0
930
END
940

FIG. 10


1000
1006
I/O DEVICES, e.g., STORAGE DEVICE
1004
MEMORY
1005
MODULE
1002
PROCESSOR

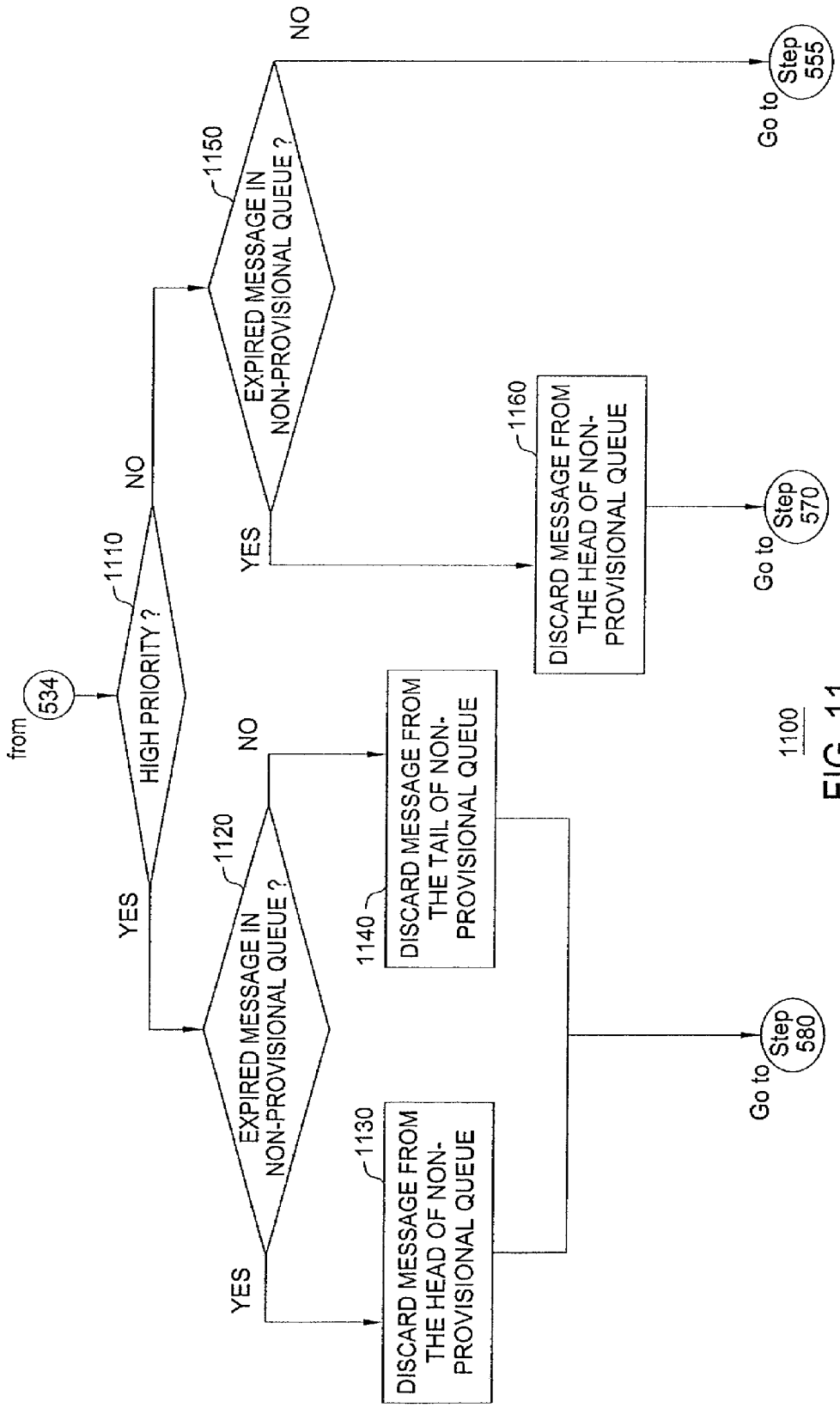

from 534
1110
HIGH PRIORITY ?
YES
NO
1120
EXPIRED MESSAGE IN NON-PROVISIONAL QUEUE ?
YES
1130
DISCARD MESSAGE FROM THE HEAD OF NON-PROVISIONAL QUEUE
NO
1140
DISCARD MESSAGE FROM THE TAIL OF NON-PROVISIONAL QUEUE
Go to Step 580
1150
EXPIRED MESSAGE IN NON-PROVISIONAL QUEUE ?
YES
1160
DISCARD MESSAGE FROM THE HEAD OF NON-PROVISIONAL QUEUE
Go to Step 570
NO
Go to Step 555
1100

FIG. 11

METHOD AND APPARATUS FOR PRIORITIZING VOICE OVER INTERNET PROTOCOL SIGNALING MESSAGES

This application is a continuation of U.S. patent application Ser. No. 12/210,701, filed Sep. 15, 2008, which is currently allowed, and is herein incorporated by reference in its entirety.

The present invention relates generally to communication networks and, more particularly, to a method and apparatus for prioritizing signaling messages in communication networks, e.g., packet networks such as Voice over Internet Protocol (VoIP) networks.

BACKGROUND OF THE INVENTION

In a well engineered network, signaling servers will not be overloaded under the normal operation mode. However, overload will occur if there are failures in the network or if there are significant increases in traffic load beyond the engineered loads. This can be the results of a disaster or mass calling due to a popular event such as reality TV shows, etc. In the former scenario, if the failures affect some signaling servers, the surviving servers will have to handle extra calls attempts. In the latter scenario, call attempts are generated so that call volumes are significantly higher than under normal operating mode.

SUMMARY OF THE INVENTION

In one embodiment, the present invention enables prioritization of signaling messages in a communication network. For example, the method receives at least one signaling message, and classifies each of the at least one signaling message. The method schedules each of the at least one signaling message for processing, and discards selectively one or more signaling messages that have been scheduled under an overload condition.

BRIEF DESCRIPTION OF THE DRAWINGS

The teaching of the present invention can be readily understood by considering the following detailed description in conjunction with the accompanying drawings, in which:

FIG. **1** illustrates an exemplary Voice over Internet Protocol (VoIP) network related to the present invention;

FIG. **2** illustrates an example signaling flow for a call setup using SIP signaling messages in a VoIP network of the present invention;

FIG. **3** illustrates an example two tier with three classes classification structure in a VoIP network of the present invention;

FIG. **4** illustrates a flowchart of a method for creating a call signaling message by a User Agent Client (UAC) in a packet network, e.g., a VoIP network, of the present invention;

FIG. **5** illustrates a flowchart of a method for processing the arrival of a signaling message at a User Agent Server (UAS) in a packet network, e.g., a VoIP network, of the present invention;

FIG. **6** illustrates a flowchart of a method for serving signaling messages in signaling message queues at a User Agent Server (UAS) in a packet network, e.g., a VoIP network, of the present invention;

FIG. **7** illustrates an example message scheduler in a VoIP network of the present invention;

FIG. **8** illustrates a flowchart of a method for discarding signaling message from a non-provisional queue using head of queue drop trigger in a packet network, e.g., a VoIP network, of the present invention;

FIG. **9** illustrates a flowchart of a method for resetting the dropping counter parameter in a packet network, e.g., a VoIP network, of the present invention;

FIG. **10** illustrates a high level block diagram of a general purpose computer suitable for use in performing the functions described herein; and

FIG. **11** illustrates a flowchart of a method for discarding signaling message from a non-provisional queue using time expiration in a packet network, e.g., a VoIP network, of the present invention.

To facilitate understanding, identical reference numerals have been used, where possible, to designate identical elements that are common to the figures.

DETAILED DESCRIPTION

To better understand the present invention, FIG. **1** illustrates a communication architecture **100** having an example network, e.g., a packet network such as a VoIP network related to the present invention. Exemplary packet networks include internet protocol (IP) networks, asynchronous transfer mode (ATM) networks, frame-relay networks, and the like. An IP network is broadly defined as a network that uses Internet Protocol to exchange data packets. Thus, a VoIP network, a SoIP (Service over Internet Protocol) network, or an IMS (IP Multimedia Subsystem) network is built on an IP network.

In one embodiment, the VoIP network may comprise various types of customer endpoint devices connected via various types of access networks to a carrier (e.g., a service provider) VoIP core infrastructure over an Internet Protocol/Multi-Protocol Label Switching (IP/MPLS) based core backbone network. Broadly defined, a VoIP network is a network that is capable of carrying voice signals as packetized data over an IP network. The present invention is described below in the context of an illustrative VoIP network. However, the present invention should not be interpreted to be limited by this particular illustrative architecture.

The customer endpoint devices can be either Time Division Multiplexing (TDM) based or IP based. TDM based customer endpoint devices **122**, **123**, **134**, and **135** typically comprise of TDM phones or Private Branch Exchange (PBX). IP based customer endpoint devices **144** and **145** typically comprise IP phones or IP PBX. The Terminal Adaptors (TA)/VoIP Gateway (Gateway)**132** and **133** are used to provide necessary interworking functions between TDM customer endpoint devices, such as analog phones, and packet based access network technologies, such as Digital Subscriber Loop (DSL) or Cable broadband access networks. TDM based customer endpoint devices access VoIP services by using either a Public Switched Telephone Network (PSTN) **120**, **121** or a broadband access network via a TA/Gateway **132** or **133**. IP based customer endpoint devices access VoIP services by using a Local Area Network (LAN) **140** and **141** with a router **142** and **143**, respectively.

The access networks can be either TDM or packet based. A TDM PSTN **120** or **121** is used to support TDM customer endpoint devices connected via traditional phone lines. A packet based access network, such as Frame Relay, ATM, Ethernet or IP, is used to support IP based customer endpoint devices via a customer LAN, e.g., **140** with a router **142**. A packet based access network **130** or **131**, such as DSL or Cable, when used together with a TA/Gateway **132** or **133**, is used to support TDM based customer endpoint devices.

The core VoIP infrastructure comprises several VoIP components, such as the Border Element (BE) **112** and **113**, the Call Control Element (CCE) **111**, VoIP related Application Servers (AS) **114**, and Media Server (MS) **115**. The BE resides at the edge of the VoIP core infrastructure and interfaces with customers endpoints over various types of access networks. A BE is typically implemented as a Media Gateway or a Session Border Controller and performs signaling, media control, security, and call admission control and related functions. The CCE resides within the VoIP infrastructure and is connected to the BEs using the Session Initiation Protocol (SIP) over the underlying IP/MPLS based core backbone network **110**. The CCE is typically implemented as a Media Gateway Controller, a softswitch, or a Call Session Control Function in an IMS network and performs network wide call control related functions as well as interacts with the appropriate VoIP service related servers when necessary. The CCE functions is a signaling endpoint for all call legs between all BEs and the CCE. The CCE may need to interact with various VoIP related Application Servers (AS) in order to complete a call that require certain service specific features, e.g. call waiting, call forwarding, voice mail, etc.

Calls that originate or terminate in a different carrier can be handled through the PSTN **120** and **121** or the Partner IP Carrier **160** interconnections. Originating or terminating TDM calls can be handled via existing PSTN interconnections to the other carrier. Originating or terminating VoIP calls can be handled via the Partner IP carrier interface **160** to the other carrier.

In order to illustrate how the different components operate to support a VoIP call, the following call scenario is used to illustrate how a VoIP call is setup between two customer endpoints. A customer using IP device **144** at location A places a call to another customer at location Z using TDM device **135**. During the call setup, a setup signaling message is sent from IP device **144**, through the LAN **140**, the Router **142**, and the associated packet based access network, to BE **112**. BE **112** will then send a setup signaling message, such as a SIP-INVITE message if SIP is used, to CCE **111**. CCE **111** processes the called party information and queries the necessary VoIP service related application server **114** to obtain the information to complete this call. In one embodiment, the Application Server (AS) functions as a SIP back-to-back user agent. If BE **113** needs to be involved in completing the call, then the CCE **111** sends another call setup message, such as a SIP-INVITE message if SIP is used, to BE **113**. Upon receiving the call setup message, BE **113** forwards the call setup message, via broadband network **131**, to TA/Gateway **133**. TA/Gateway **133** then identifies the appropriate TDM device **135** and rings that device.

Once the call is accepted at location Z by the called party, a call acknowledgement signaling message, such as a SIP 200 OK response message if SIP is used, is sent in the reverse direction back to the CCE **111**. After the CCE **111** receives the call acknowledgement message, it will then send a call acknowledgement signaling message, such as a SIP 200 OK response message if SIP is used, towards the calling party. In addition, the CCE **111** also provides the necessary information of the call to both BE **112** and BE **113** so that the media exchange can proceed directly between BE **112** and BE **113**. The call signaling path **150** and the call media path **151** are illustratively shown in FIG. **1**. Note that the call signaling path and the call media path are different because once a call has been setup up between two endpoints, the CCE **111** does not need to be in the media path for actual direct media exchange.

Note that a customer in location A using any endpoint device type with its associated access network type can communicate with another customer in location Z using any endpoint device type with its associated network type as well. For instance, a customer at location A using IP customer endpoint device **144** with packet based access network **140** can call another customer at location Z using TDM endpoint device **123** with PSTN access network **121**. The BEs **112** and **113** are responsible for the necessary signaling protocol translation, e.g., SS7 to and from SIP, and media format conversion, such as TDM voice format to and from IP based packet voice format.

Media Servers (MS) **115** are special servers that typically handle and terminate media streams, and to provide services such as announcements, teleconference bridges, transcoding, and Interactive Voice Response (IVR) messages for VoIP service applications.

VoIP services have proliferated in recent years due to rapid advance in technology and market demands. Service providers are aggressively looking for ways to offer VoIP to customers via various Quality of Service (QoS) mechanisms. However, the existing discussions on providing QoS to VoIP traffic primarily focus on the VoIP call media path such as how to ensuring voice quality after calls are accepted. Little has been done on providing QoS treatment for VoIP signaling messages, such as SIP signaling messages, especially at SIP servers that are involved in setting up VoIP calls. In one embodiment of the present invention, CCE **111** shown in FIG. **1** is a SIP server.

Providing QoS treatment for VoIP signaling messages at SIP servers is beneficial when SIP servers are overloaded. In a well engineered network, SIP servers will not be overloaded under the normal operation mode. However, overload will occur if there are failures in the network or if there are significant increases in traffic load beyond the engineered loads.

When SIP servers are overloaded, signaling messages will be dropped at the SIP servers. The dropped signaling messages may be signaling messages that need to be treated with higher priority. For example, calls from callers such as government emergency control agents that manage or assist in the rescue efforts in a disaster, should be handled with priority. To prevent signaling message dropping of important calls, QoS features have to be implemented at SIP servers so that important messages will always be processed even under overload conditions.

The present invention enables differential QoS treatments of various signaling messages at signaling servers. In one embodiment, the present invention employs three components: classification of signaling messages, scheduling of signaling messages for processing, and selective discarding of signaling messages under overload conditions, to support differential QoS treatments of signaling messages. Signaling message are first classified to different priority levels and then scheduled to be processed and, if necessary, when memory space runs out, discarded by a signaling server according to their classified priority levels.

FIG. **2** provides an example signaling flow **200** for a call setup using SIP signaling messages. In FIG. **2**, endpoint **210** initiates a call setup request towards endpoint **220** via a proxy **230** and a proxy **240** as shown. In this example, endpoint **210** and endpoint **220** are User Agent Clients (UAC) and proxy **230** and proxy **240** are User Agent Server (UAS) for SIP signaling purposes. User Agent Client (UAC) is a logical entity that creates a new SIP request and then sends it to the network. User Agent Server (UAS) is a logical entity that generates a response to a SIP request. The response accepts, rejects, or redirects the request.

For instance, SIP INVITE message F**1** is generated and sent by endpoint **210** to proxy **230**. Upon receiving INVITE

message F1, proxy **230**, acting as both a UAC and a UAS, generates and forwards an INVITE message F2 to proxy **240**. In addition, proxy **230**, acting both as a UAC and a UAS, also responds to INVITE message F1 by generating and sending the SIP 100 TRYING message F3 to endpoint **210**.

Upon receiving INVITE message F2, proxy **240** generates and forwards INVITE message F4 to endpoint **220** in response to INVITE message F2. In addition, proxy **240** also generates a SIP 100 TRYING message F5 to proxy **230** in response to INVITE message F2. The rest of the signaling message flows can be similarly interpreted. Note that once a call is setup between the two endpoints, the endpoints can then signal each other directly without going through the proxies. For instance, SIP ACK message F12, BYE message F13, and 200 OK message F14 are such directly exchanged signaling message between endpoint **210** and endpoint **220**. The media session flow represents the call media path and flow between endpoint **210** and endpoint **220**.

In one embodiment, the classification component of the present invention decides which message gets higher priority than others. For example, a two-tier classification approach is introduced. In the first tier, calls are categorized into two classes, a high priority class and a low priority class. The high priority class comprises signaling messages that are more important than those of low priority class. The criteria of such categorization can be flexibly configured by individual service providers. For instance, calls originated from government emergency agents (e.g., government employees (federal, state, city, county, etc.), medical personnel, utility employees, and so on) can be classified as the high priority class. In the second tier, signaling messages from the low priority class of calls are further classified into two classes based upon the nature of the messages. For instance, the SIP call INVITE and BYE messages can be configured to have higher priority than other SIP signaling messages, such as 1xx messages comprising provisional responses. Again, the actual classification criteria within the lower priority class can be flexibly configured by individual service providers. SIP messages of different classes will then be queued separately so that they can be served in a differential manner.

This illustrative two tier classification structure is based upon both the call type and the SIP signaling message type. Though one can define as many classes as it is needed, in one embodiment, a three class structure using this two tier classification is used:

- Class 1: all SIP messages associated with high priority calls;
- Class 2: important SIP signaling messages from low priority calls, also know as non-provisional signaling messages; and
- Class 3: non-important SIP signaling messages from lower priority calls, also know as provisional messages.

Note that the number of high priority calls should be restricted to a small percentage of all call types. Therefore, the amount of signaling message traffic of high priority calls shall be engineered to never overload any signaling servers in a VoIP network.

FIG. **3** illustrates an example **300** of the two tier with three classes signaling message classification scheme in a VoIP network of the present invention. In FIG. **3**, all incoming signaling messages arrive first into queue **310**. Then these messages are parsed and pre-processed by module **320**. The IP header Differentiated Service Code Point (DSCP) field is used to determine the classes to which incoming signaling message belong. Once the classification is performed, signaling message are then queued into the appropriate classes, e.g., queue **330** for storing high priority messages, queue **340** for storing non-provisional messages, and queue **350** for storing provisional messages. Queued signaling messages in these queues are then processed by module **360** according to the present invention described herein.

Since SIP is an application layer protocol employing IP networks for transmissions, the actual SIP messages are encapsulated within IP packets while traversing through IP networks. When IP packets carrying SIP messages arrive at a SIP server, the SIP server needs to determine what type of SIP message is carried within an arriving IP packet first before placing it into a service queue. Thus, IP packet header needs to carry a SIP message identity. In one embodiment, the present invention uses an existing IP header field for classification purpose, such as the Differentiated Service Code Point (DSCP) field in the IP packet header. The DSCP field is an 8 bit field within the IP packet that can be used to prioritize the importance of the IP packet.

The marking for IP packets with encapsulated SIP messages is performed whenever a new SIP message is composed. For instance, when a call request arrives at a User Agent Client (UAC), a new SIP INVITE message is composed and created and the resulting IP packet is appropriately marked by the UAC. Note that some SIP messages are generated at SIP servers as responses to INVITE messages and their resulting IP packets will need to be marked accordingly before transmission as well.

Moreover, for any given call setup flow initiated at a SIP User Agent Client (UAC), there could be multiple SIP User Agent Servers (UAS) along the signaling path. Therefore, some SIP response messages will need to be forwarded by SIP servers that are not the origin of the SIP response messages. Queuing only occurs for those SIP messages that arrive at a SIP server and must wait for processing.

FIG. **4** illustrates a flowchart of a method **400** for processing a call request by a SIP User Agent Client (UAC) in VoIP network, of the present invention. Method **400** starts in step **405** and proceeds to step **410**.

In step **410**, the method receives a call request at the UAC. In step **420**, the method inspects the received call request to determine the classification of the call request. In one embodiment, the call request will be classified using the two tier and three class classification structure based on the customer type and the signaling message type. For example, the SIP INVITE signaling message for a call associated with a particular customer can be assigned to the high priority class.

In step **430**, a SIP INVITE signaling message is composed and created. In step **440**, the method encapsulates the composed SIP INVITE message within an IP packet. In step **450**, the method marks the IP packet with the appropriate classification determined in step **420**. In one embodiment, the IP packet is marked using the DSCP field in the IP packet header. The method ends in step **460**.

The scheduling of signaling message for processing component of the present invention places SIP messages arriving at a SIP server into different signaling message service queues. When IP packets with SIP messages arrive at a SIP server, the packet headers are inspected and the packets are then placed into different queues. The SIP server then processes those SIP messages according to the defined service scheduling algorithm. In one embodiment, the present invention uses a service scheduling algorithm that takes into consideration the time sensitive nature of SIP messages.

In one embodiment, the scheduling algorithm is a hybrid scheduling algorithm that combines strict priority scheduling and First In First Out (FIFO) scheduling. The strict priority scheduling means that a SIP server will process SIP messages from low priority call queues only when the high priority call

queue is empty. For all other low priority call queues, the SIP server will serve the SIP message that has the longest waiting time regardless of which low priority queue that the SIP message is in. Essentially, the SIP messages in the low priority queues are being served in FIFO fashion.

To ensure the proper order of services between two low priority queues, a message scheduler, is used to track the order of arrivals. Whenever a new message arrives, it will be time stamped. A record is entered into the message scheduler indicating the type of the message and its associated arrival time.

The reason for queuing low priority SIP messages into separate queues is to allow intelligent selective discarding of signaling messages under overload conditions.

For example, during a national disaster such as hurricane Katrina, calls from all over the world would be directed to the disaster sites, the number of call attempts will be overwhelmingly high even five or ten times higher than the normal load. Thus, the SIP servers may become the bottleneck. The existing SIP servers on the market will treat all the call attempts in a FIFO fashion. Under normal conditions, in a well engineered network, SIP messages can be handled in a timely fashion using FIFO serving discipline. However, under the overload conditions, the processing rate of a SIP server will become lower than the SIP message arrival rate, and will be accumulated in queues at the SIP server. Unfortunately, the arriving SIP messages will be discarded upon arrival when the queues reach their full capacity.

The issue associated with the FIFO treatment of all messages at SIP server is that the dropped messages can be the ones that trigger retransmission of SIP messages from originating UAC. Consequently, more messages are created to flood the already overloaded SIP servers. In a typical SIP call setup flow, it is essentially a three-way handshake session of involving SIP INVITE, SIP 200, and SIP ACK messages and a media session is terminated by a two-way handshake involving SIP BYE and SIP 200 messages. All other messages are provisional ones that will not trigger retransmission of messages toward a SIP server.

By putting different messages into different queues, one has the advantage of deciding whether to discard a low priority message already sitting in a queue so as to be able to accept the newly arrived high priority message.

Moreover, due to the time sensitive nature of non-provisional messages, serving a non-provisional message that has been sitting in the queue for too long is a waste of resources. Particularly when the volume of requests to high priority queue is unusually high, waiting time for the messages in the non-provisional queue can be significantly long. One way to overcome this problem is to discard the message from the head of the queue whenever it is necessary. It is sufficient to apply the head of queue dropping to the non-provisional queue only.

In the present invention, two mechanisms can be used to determine when a message should be dropped from the head of queue. One approach is the time expiration message discarding method, and another approach is the head drop trigger message discarding method.

In the time expiration message discarding method, a user configurable parameter, such as an expiration time, is introduced. The value of expiration time is the maximum allowable time that a non-provisional message can remain in the non-provisional queue. Whenever a message has to be deleted from the non-provisional queue, the waiting time of the head of queue will be checked first. If the waiting time is larger than the expiration time, the message will be deleted from the head of the queue rather than from the tail of the queue. Otherwise, the message form the tail of the queue will be discarded.

A detailed flowchart of the processing of an arrived signaling message is also provided in FIG. **5**. FIG. **5** illustrates a flowchart of a method **500** for processing the arrival of a signaling message at a UAS in a VoIP network of the present invention. Method **500** starts in step **505** and proceeds to step **510**.

In step **510**, the method receives a SIP signaling message via an IP network. In step **515**, the method inspects the IP header of the IP packet that is received. For example, the priority of the SIP message is extracted using information embedded in the IP packet header.

In step **520**, the method checks if the SIP server memory is completely utilized. If the SIP server memory is completely utilized, the method proceeds to step **523**; otherwise, the method proceeds to step **563**.

In step **523**, the method checks if the SIP message is a provisional signaling message. If the SIP message is a provisional signaling message, the method proceeds to step **555**; otherwise, the method proceeds to step **525**.

In step **525**, the method checks if the provisional message queue is empty. If the provisional message queue is empty, the method proceeds to step **528**; otherwise, the method proceeds to step **560**.

In step **528**, the method checks if the non-provisional message queue is empty. If the non-provisional message queue is empty, the method proceeds to step **555**; otherwise, the method proceeds to step **532**.

In step **532**, the method checks the type of discard method to be used to drop signaling messages. If the time expiration discard method is used, the method proceeds to step **534**. If the head of queue drop trigger discard method is used, the method proceeds to step **536**.

In step **534**, the method executes the time expiration discard method (as further described below in method **1100**). After the execution of step **534**, the method proceeds to step **555** if the received message is a non-provisional message and there is no time expired non-provisional message in the non-provisional message queue, or to step **570** if the received message is a non-provisional message and there is an expired message in the non-provisional message queue, or to step **580** if the received message is high priority message.

In step **536**, the method executes the head of queue drop trigger discard method described in method **800**. After the execution of step **536**, the method proceeds to step **570** if the received message is a non-provisional message or to step **580** if the received message is a high priority message.

In step **555**, the method discards the received signaling message. The method then proceeds to step **590**.

In step **560**, the method discards a message from the tail of the provisional message queue. The method then proceeds to step **563**.

In step **563**, the method checks if the received message is a high priority message. If the received message is a high priority message, the method proceeds to step **580**; otherwise, the method proceeds to step **565**.

In step **565**, the method checks if the received message is a provisional message. If the received message is a provisional message, the method proceeds to step **575**; otherwise, the method proceeds to step **570**.

In step **570**, the method places the received message into the provisional message queue. The method then proceeds to step **585**.

In step **575**, the method places the received message into the non-provisional message queue. The method then proceeds to step **585**.

In step **580**, the method places the received message into the high priority message queue. The method then proceeds to step **585**.

In step **585**, the method updates the message scheduler accordingly. The method then ends in step **590**.

FIG. **11** illustrates a flowchart of a method **1100** for discarding signaling message from a non-provisional queue using time expiration in a VoIP network, of the present invention. Method **1100** starts when step **534** in method **500** is reached and the method proceeds to step **1110**.

In step **1110**, the method checks if the received message is a high priority message. If the received message is a high priority message, the method proceeds to step **1120**; otherwise, the method proceeds to step **1150**.

In step **1120**, the method checks if the head of queue message in the non-provisional message queue has exceeded the expiration timer threshold. If the head of queue message in the non-provisional message queue has exceeded the expiration timer threshold, the method proceeds to step **1130**; otherwise, the method proceeds to step **1140**.

In step **1130**, the method discards the head of queue message from the non-provisional message queue. The method then proceeds to step **580** in method **500**.

In step **1140**, the method discards the tail of queue message from the non-provisional message queue. The method then proceeds to step **580** in method **500**.

In step **1150**, the method checks if the head of queue message in the non-provisional message queue has exceeded the expiration timer threshold. If the head of queue message in the non-provisional message queue has exceeded the expiration timer threshold, the method proceeds to step **1160**; otherwise, the method proceeds to step **555** in method **500**.

In step **1160**, the method discards the head of queue message from the non-provisional message queue. The method then proceeds to step **570** in method **500**.

In the head drop trigger message discarding method, a user configurable parameter, such as a head of queue drop trigger, is introduced. Whenever a signaling message in the non-provisional queue is deleted due to high priority message arrivals, the server is spending more time on the high priority queue. It indicates that the messages in the non-provisional queue have to wait longer to be served. If a significant number of messages are deleted from the non-provisional queue since the last non-provisional message was served, there is a very high likelihood that the signaling message waiting at the head of the non-provisional queue is too old to be meaningful. Thus, if the number of deleted messages from non-provisional queue exceeds the user configured parameter of the head of queue drop trigger, the head of queue discard will be performed if a non-provisional message has to be dropped.

To implement the head drop trigger message discarding method, the concept of dropping zone is introduced. There are two dropping zones, the tail of queue dropping zone and the head of queue dropping zone. Furthermore, a dropping counter to keep track of number of consecutive tail of queue drops from the non-provisional queue is introduced. Consecutive dropping is defined to be the number of discard for non-provisional signaling messages since the last non-provisional message was served. When the dropping counter exceeds the head of queue drop trigger, the dropping zone is the head of queue dropping zone, otherwise it is the tail of queue dropping zone. If the dropping zone is in the tail of queue dropping zone, the dropping counter is reset to zero whenever a non-provisional message is served or scheduled. If it is in the head of queue dropping zone, the dropping counter is decremented by one whenever a head of queue discard occurs until it is zero.

FIG. **8** illustrates a flowchart of a method **800** for discarding a signaling message from a non-provisional queue using head of queue drop trigger in a VoIP network of the present invention. Method **800** starts when step **536** in method **500** is reached and the method proceeds to step **807**.

In step **807**, the method checks to verify if the method has previously been initialized already. If the method has previously been initialized already, the method proceeds to step **813**; otherwise, the method proceeds to step **810**.

In step **810**, the method initializes the dropping counter parameter to 0 and the dropping zone to tail of queue dropping when an initial signaling message arrives. The dropping counter parameter keeps track of the number of consecutive tail of queue drops from the non-provisional queue. The dropping zone parameter can either be the tail of queue dropping zone or the head of queue dropping zone, indicating signaling message shall be discarded from the tail of queue or the head of queue, respectively.

In step **813**, the method determines if the received signaling message is a high priority message. If the received signaling message is a high priority message, the method proceeds to step **815**; otherwise, the method proceeds to step **860**.

In step **815**, the method checks if the dropping zone parameter is set to the tail of queue dropping zone. If the dropping zone parameter is set to the tail of queue dropping zone, the method proceeds to step **820**; otherwise, the method proceeds to step **840**.

In step **820**, the method increments the dropping counter parameter by 1.

In step **825**, the method checks if the dropping counter value exceeds the head of queue drop trigger parameter. Note that the head of queue drop trigger parameter is a user configurable parameter set by the user. If the dropping counter value exceeds the head of queue drop trigger parameter, the method proceeds to step **830**; otherwise, the method proceeds to step **835**.

In step **830**, the method sets the dropping zone parameter to the head of queue dropping zone.

In step **835**, the method discards a non-provisional signaling message from the tail of the non-provisional queue. The method then proceeds to step **580** in method **500**.

In step **840**, the method decrements the dropping counter parameter by 1.

In step **845**, the method checks if the dropping counter parameter value is 0. If the dropping counter parameter value is 0, the method proceeds to step **850**; otherwise, the method proceeds to step **855**.

In step **850**, the method sets the dropping zone parameter to the tail of queue dropping zone. In step **855**, the method discards a non-provisional signaling message from the head of the non-provisional queue. The method then proceeds to step **580** in method **500**.

In step **860**, the method discards a non-provisional signaling message from the tail of the non-provisional queue. The method then proceeds to step **570** in method **500**.

FIG. **9** illustrates a flowchart of a method **900** for resetting the dropping counter parameter in a VoIP network of the present invention. Method **900** starts in step **905** and proceeds to step **907**.

In step **907**, a signaling message is served by a signaling server. In step **910**, the method checks if the next signaling message to be processed is a message from the non-provisional queue. If the next signaling message to be processed is a message from the non-provisional queue, the method proceeds to step **920**; otherwise, the method proceeds to step **940**.

In step **920**, the method checks if the dropping zone parameter is set to the tail of queue dropping zone. If the dropping

zone parameter is set to the tail of queue dropping zone, the method proceeds to step **930**; otherwise, the method proceeds to step **940**.

In step **930**, the method resets the dropping counter parameter to a value of 0. The method ends in step **940**.

Compared to threshold based queue management schemes, another advantage of separate queues for SIP messages is to maximize utilization of the memory space in a SIP server. The four associated procedures that are performed by a SIP server will be described as follows.

In a first procedure, in order to facilitate the differential treatment of SIP messages in a SIP server, the classification rules are programmed into the SIP server. A user interface is used by service providers to define and configure rules within the server according to service needs.

In a second procedure, a message scheduler is used to track the order of arrival of SIP signaling messages, including both provisional and non-provisional signaling messages, which do not belong to high priority queue. Whenever such a new SIP signaling message arrives, the arrival order is recorded into the message scheduler. Whenever a message is processed, the associated record is eliminated from the message scheduler.

FIG. **7** illustrates an example message scheduler **700** in a VoIP network of the present invention. State **710** illustrates the initial state of the message scheduler before a signaling message, either provisional or non-provisional, is served by a SIP signaling server. State **720** illustrates the state of the message scheduler after the head of queue provisional signaling message is served. State **730** illustrates the state of the message scheduler after a provisional signaling message is received and inserted into the tail of queue.

In a third procedure, a SIP signaling message that arrives at a User Agent Server (UAS) is processed. The pseudo code of the processing of an arrived signaling message is given below.

```
If it is from a higher priority source,
    If the memory space is not full
Put it into high priority queue
    Else
        Discard a SIP message from provisional queue
        Put the newly arrived message into high priority queue
Else
    Update Message Scheduler - enter its record
    If it is not a provisional message (determined by rules)
        If the memory space is not full
            Put it into a non-provisional queue
         Else
            Discard a provisional message from provisional
queue
Put the newly arrived message into non-provisional queue
    Else
        If the memory space is not full
Put it into a provisional queue
            Else
                Discard the newly arrived SIP message
```

In a fourth procedure, the signaling messages in the various message queues are served by the UAS. The pseudo code of this processing is given below.

```
If high priority queue is not empty
        Process a SIP message from the high priority queue
Else
        Check the Message Scheduler and select the SIP message
accordingly
        Process the chosen SIP message
        Update the Message Scheduler - eliminate its record
        Ready to process next message
```

A detailed flowchart of the serving of signaling messages in various signaling message queues is provided in FIG. **6**. FIG. **6** illustrates a flowchart of a method **600** for serving signaling messages in signaling message queues at a UAS in a VoIP network of the present invention. Method **600** starts in step **605** and proceeds to step **610**.

In step **610**, the method awaits to serve the next SIP message in signaling message queues after a previous message has been served.

In step **615**, the method checks if the high priority message queue is empty. If the high priority message queue is empty, the method proceeds to step **620**; otherwise, the method proceeds to step **645**.

In step **620**, the method checks if the message scheduler is empty. If the message scheduler is empty, the method proceeds back to step **610**; otherwise, the method proceeds to step **625**.

In step **625**, the method checks if the next signaling message to be served is a message from the provisional message queue according to the message scheduler. If the next signaling message to be served is a message from the provisional message queue according to the message scheduler, the method proceeds to step **630**; otherwise, the method proceeds to step **635**.

In step **630**, the method processes the signaling message from the head of queue of the provisional message queue. The method then proceeds to step **640**.

In step **635**, the method processes the signaling message from the head of queue of the non-provisional message queue. The method then proceeds to step **640**.

In step **640**, the method updates the message scheduler accordingly after a signaling message has been processed. The method then proceeds back to step **610**.

In step **645**, the method processes a signaling message from the head of queue of the high priority message queue. The method then proceeds to step **610**.

This fourth procedure guarantees call setup of high priority calls. While the third procedure effectively serves as an overload control mechanism to minimize the SIP message retransmissions due to loss of non-provisional messages so as to increase the resource utilization as well as the successful call setup rate.

It should be noted that although not specifically specified, one or more steps of methods **400**, **500**, **600**, **800**, **900**, and **1100** may include a storing, displaying and/or outputting step as required for a particular application. In other words, any data, records, fields, and/or intermediate results discussed in the method can be stored, displayed and/or outputted to another device as required for a particular application. Furthermore, steps or blocks in FIGS. **4**, **5**, **6**, **8**, **9**, and **11** that recite a determining operation or involve a decision, do not necessarily require that both branches of the determining operation be practiced. In other words, one of the branches of the determining operation can be deemed as an optional step.

FIG. **10** depicts a high level block diagram of a general purpose computer suitable for use in performing the functions described herein. As depicted in FIG. **10**, the system **1000** comprises a process serving signaling messages in signaling message queues at a UAS or element **1002** (e.g., a CPU), a memory **1004**, e.g., random access memory (RAM) and/or read only memory (ROM), a module **1005** for prioritizing VoIP signaling messages, and various input/output devices **1006** (e.g., storage devices, including but not limited to, a tape drive, a floppy drive, a hard disk drive or a compact disk drive, a receiver, a transmitter, a speaker, a display, a speech synthesizer, an output port, and a user input device (such as a keyboard, a keypad, a mouse, and the like)).

It should be noted that the present invention can be implemented in software and/or in a combination of software and hardware, e.g., using application specific integrated circuits (ASIC), a general purpose computer or any other hardware equivalents. In one embodiment, the present module or process **1005** for prioritizing VoIP signaling messages can be loaded into memory **1004** and executed by processor **1002** to implement the functions as discussed above. As such, the present process **1005** for prioritizing VoIP signaling messages (including associated data structures) of the present invention can be stored on a computer readable medium or carrier, e.g., RAM memory, magnetic or optical drive or diskette and the like.

While various embodiments have been described above, it should be understood that they have been presented by way of example only, and not limitation. Thus, the breadth and scope of a preferred embodiment should not be limited by any of the above-described exemplary embodiments, but should be defined only in accordance with the following claims and their equivalents.

What is claimed is:

1. A method for discarding a signaling message in a communication network, comprising:

receiving, by a processor, the signaling message;

classifying, by the processor, the signaling message;

scheduling, by the processor, the signaling message for processing; and

discarding, by the processor, selectively the signaling message that has been scheduled under an overload condition, wherein the discarding comprises:

discarding the signaling message that has been scheduled from a tail of a provisional message queue when the provisional message queue is not empty, when a high priority message or a non-provisional signaling message is newly received and when the signaling message is a provisional signaling message; and

discarding the signaling message that has been scheduled from a non-provisional message queue when the provisional message queue is empty and the non-provisional message queue is not empty, when a high priority message or a non-provisional signaling message is newly received and when the signaling message is a non-provisional signaling message, wherein the discarding the signaling message that has been scheduled from the non-provisional message queue comprises a head drop trigger discard method, wherein the head drop trigger discard method comprises:

discarding the signaling message from a head of the non-provisional message queue when a dropping zone is set to a head dropping zone; and

discarding the signaling message from a tail of the non-provisional message queue when the dropping zone is set to a tail dropping zone;

discarding, by the processor, a newly received signaling message under the overload condition when the newly received signaling message is a provisional signaling message; and

discarding, by the processor, the newly received signaling message under the overload condition when both of the provisional message queue and the non-provisional message queue are empty.

2. The method of claim **1**, wherein the communication network is a voice over internet protocol network.

3. The method of claim **1**, wherein the signaling message is classified into one of: a high priority, a non-provisional priority, and a provisional priority.

4. The method of claim **3**, wherein the signaling message is classified based on a customer type and a signaling message type by a user agent client.

5. The method of claim **3**, wherein the signaling message is classified based on a customer type and a signaling message type by a user agent server.

6. The method of claim **3**, wherein the scheduling comprises:

placing the signaling message into one of a plurality of class processing queues;

scheduling all signaling messages classified as high priority to be processed before signaling messages classified as non-provisional priority, or as provisional priority; and

scheduling signaling messages classified as non-provisional priority, or as provisional priority, using a first in first out order.

7. The method of claim **1**, wherein the signaling messages are session initiation protocol signaling messages.

8. A non-transitory computer-readable medium storing a plurality of instructions which, when executed by a processor, cause the processor to perform operations for discarding a signaling message in a communication network, the operations comprising:

receiving the signaling message;

classifying the signaling message;

scheduling the signaling message for processing;

discarding selectively the signaling message that has been scheduled under an overload condition, wherein the discarding comprises:

discarding the signaling message that has been scheduled from a tail of a provisional message queue when the provisional message queue is not empty, when a high priority message or a non-provisional signaling message is newly received and when the signaling message is a provisional signaling message; and

discarding the signaling message that has been scheduled from a non-provisional message queue when the provisional message queue is empty and the non-provisional message queue is not empty, when a high priority message or a non-provisional signaling message is newly received and when the signaling message is a non-provisional signaling message, wherein the discarding the signaling message that has been scheduled from the non-provisional message queue comprises a head drop trigger discard method, wherein the head drop trigger discard method comprises:

discarding the signaling message from a head of the non-provisional message queue when a dropping zone is set to a head dropping zone; and

discarding the signaling message from a tail of the non-provisional message queue when the dropping zone is set to a tail dropping zone;

discarding a newly received signaling message under the overload condition when the newly received signaling message is a provisional signaling message; and

discarding the newly received signaling message under the overload condition when both of the provisional message queue and the non-provisional message queue are empty.

9. The non-transitory computer-readable medium of claim **8**, wherein the communication network is a voice over Internet protocol network.

10. The non-transitory computer-readable medium of claim 8, wherein the signaling message is classified into one of: a high priority, a non-provisional priority, and a provisional priority.

11. The non-transitory computer-readable medium of claim 10, wherein the signaling message is classified based on a customer type and a signaling message type by a user agent client.

12. The non-transitory computer-readable medium of claim 10, wherein the signaling message is classified based on a customer type and a signaling message type by a user agent server.

13. The non-transitory computer-readable medium of claim 10, wherein the scheduling comprises:

placing the signaling message into one of a plurality of class processing queues;

scheduling all signaling messages classified as high priority to be processed before signaling messages classified as non-provisional priority, or as provisional priority; and

scheduling signaling messages classified as non-provisional priority, or as provisional priority, using a first in first out order.

14. An apparatus for discarding a signaling message in a communication network, comprising:

a processor; and

a computer-readable medium storing a plurality of instructions which, when executed by the processor, cause the processor to perform operations, the operations comprising:

receiving the signaling message;

classifying the signaling message;

scheduling the signaling message for processing;

discarding selectively the signaling message that has been scheduled under an overload condition, wherein the discarding comprises:

discarding the signaling message that has been scheduled from a tail of a provisional message queue when the provisional message queue is not empty, when a high priority message or a non-provisional signaling message is newly received and when the signaling message is a provisional signaling message; and

discarding the signaling message that has been scheduled from a non-provisional message queue when the provisional message queue is empty and the non-provisional message queue is not empty, when a high priority message or a non-provisional signaling message is newly received and when the signaling message is a non-provisional signaling message, wherein the discarding the signaling message that has been scheduled from the non-provisional message queue comprises a head drop trigger discard method, wherein the head drop trigger discard method comprises:

discarding the signaling message from a head of the non-provisional message queue when a dropping zone is set to a head dropping zone; and

discarding the signaling message from a tail of the non-provisional message queue when the dropping zone is set to a tail dropping zone;

discarding a newly received signaling message under the overload condition when the newly received signaling message is a provisional signaling message; and

discarding the newly received signaling message under the overload condition when both of the provisional message queue and the non-provisional message queue are empty.

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